Speech Compression Using Wavelet Packet Best Tree Encoding (BTE)

Amr M. Gody, Saby A. Abdelwahab, Tamer M. Barakat and Mohamed Y. Mohamed

Abstract—Speech compression is one area of digital signal processing that focusing on reducing the bit rate of the speech signal for transmission or storage without significant loss of quality. This paper presents a new design feature for speech Compression using Wavelet Packet Transform and Linear prediction Coding. The proposed algorithm uses best tree decomposition of wavelet packets, which develop features afterward compressed signal is compressed by Linear prediction Coding. The 4 points encoded vector is a full of information just like the original best tree’s structure. The implied scoring system makes BTE suitable for compression problems and we reach at 12 compression ratio. The performance of speech signal is measure on the basis of signal to noise ratio (SNR), mean square error (MSE) and peak signal to noise ratio (PSNR).

Keywords—Speech Compression; Wavelet Packet Transform; BTE; LPC

I. INTRODUCTION

Speech compression has been and still is a major issue in the area of digital speech processing. Speech compression is the act of transforming the speech signal at hand, to a more compact form, which can then be transmitted with a considerably smaller memory. The motivation behind this is the fact that access to unlimited amount of bandwidth is not possible. Therefore, there is a need to code and compress speech signals. Speech compression is required in long distance communication, high quality speech storage, and message encryption.

For example, in digital cellular technology many users need to share the same frequency bandwidth. Utilizing speech compression makes it possible for more users to share the available system. Another example where speech compression is needed is in digital voice storage. For a fixed amount of available memory, compression makes it possible to store longer messages [1-4].

Speech compression is a lossy type of coding, which means that the output signal does not exactly sound like the input. The input and output signal could be distinguished to be different. Speech compression tries to code the speech in a perceptually lossless way. This means that even though the input and output signals are not mathematically equivalent, the sound at the output is the same as the input. This type of compression is used in applications for audio storage, broadcasting, and Internet streaming [5]. Several techniques of speech compression such as Linear Predictive Coding (LPC), Waveform Coding and Sub-band Coding are existed.

The objective of this paper is to introduce new features for speech signal. Features are developed from the wavelet packets best tree decomposition of speech signal. This research aims to explain the proposed features in details. Also, it targets to introduce the benefits of using the proposed feature in speech compression problems.

This paper is outlined as follows: In section II, the concept and motivation of wavelet transform are reviewed. Section III presents feature extraction. Section IV gives linear prediction coding. Finally, the results and the concluding remarks are given in section V and VI.

II. WAVELET TRANSFORM

Wavelets are short duration waveforms that can express any function by scaling and shifting of certain mother signal that is called mother wavelet [6]. Wavelet algorithm is acting as a filter banks on the input signal. The output of the filter banks are the wavelet signal’s amplitudes.

Fig 1. Sin Wave is used for Fourier Representation of the Signal While Wavelet Function is used in Wavelet Representation for Daubechies 10 Pointes Filter. Sin Wave is Infinite in Time but Finite in Frequency Domain While Wavelet is Finite in Both Time and Frequency Domains.

Figure 1 indicates a very important property of wavelet function. Wavelet function is a finite in time. It is also finite in frequency [7]. This is not the case of "Sine" basis functions (harmonic functions) used for Fourier analysis. All derived wavelets are orthogonal. This makes each wavelet acts as an identifier of the signal in a certain band. Figure 2 gives a brief
comparison between different possible spaces to express certain function [8].

Wavelet packets are an extension to wavelet transform. It includes the high frequency parts in the analysis for more signal resolution of the frequency spectrum as shown in figure 3.

To simplify the subject, let us discuss Fourier series as a signal representation tool.

\[ f(x) = a_0 + \sum_{n=1}^{\infty} (a_n \cos \frac{n\pi x}{L} + b_n \sin \frac{n\pi x}{L}) \] (1)

Equation 1 indicates the Fourier series representation of function \( f(x) \). By the same approach, \( f(x) \) may be expressed using wavelet packets as in equation 2.

\[ f(x) = \sum_j \sum_{n=0}^{2^{j-1}} b_{jn} W_{j,n}(x) \] (2)

Where \( b \) is wavelet coefficients and \( W \) is wavelet packet. Let us start with the two filters of length \( 2N \), where \( h(n) \) and \( g(n) \), corresponding to the wavelet filters.

\[ W_{2n}(x) = \sqrt{2} \sum_{k=0}^{2N-1} h(k) W_n(2x - k) \] (3)

\[ W_{2n+1}(x) = \sqrt{2} \sum_{k=0}^{2N-1} g(k) W_n(2x - k) \] (4)

g(k) and h(k) are filter banks. Where:

\[ W_0(x) = \phi(x) \] is called the scaling function.

\[ W_{j,n}(x) = w_n(2^{-j} x - k) \] (5)

\( K \) is not a dynamic parameter after the decompositon of the signal rather it is a constant value for each wavelet packet \( W \). This makes it much better to abstract (5) as:

\[ W_{j,n} = w_n(2^{-j} x - k) \] (6)

Hence:

\[ W_{0,0}(x) = \Phi(x - k) \] (7)

\[ W_{1,1}(x) = \Psi\left(\frac{x}{2} - k\right) \] (8)

The idea is explained by figure 4. Scaling "\( \Phi \)" and wavelet "\( \Psi \)" functions are used to generate \( W \) functions that cover all the frequency-scale space. The parameter \( k \) is used to indicate the time location of certain \( W \) function. \( K \) is chosen to best fit the original function to be expressed by wavelet packets while the scaling and wavelet functions are designed such that all \( W \) functions to be orthogonal.

In this section the process of feature extraction using Best Tree 4 point Encoded features (BTE) will be explained. Wavelet packets process is very similar to filter banks. Both of them are filter banks in nature. The wavelet packets method is a generalization of wavelet decomposition that offers a richer signal analysis. Wavelet packet atoms are waveforms indexed by three naturally interpreted parameters: position, scale (as in wavelet decomposition), and frequency. For a given orthogonal wavelet function, we generate a library of bases called wavelet packet bases. Each of these bases offers a particular way of coding signals, preserving global energy, and reconstructing exact features. The wavelet packets can be used for numerous
expansions of a given signal. We then select the most suitable decomposition of a given signal with respect to an entropy-based criterion [12].

The first step in BTE is to align the neighboring bands. This is very important for a good scoring process. Scoring process tries to score adjacent bands in such that minimizing the distance. For our case of best tree by Matlab, adjacent bands are indexed not in sequence.

Fig 5. Wavelet Packet Tree Analysis Chart to Figure Out Adjacent Bands.

The objective is to remap node indices in such that adjacent node indices lay in adjacent frequency bands. To explain this subject considers the following table that represents the indices in a typical wavelet packet tree for 4-levels decomposition. Figure 5 represents band indexes in Matlab wavelet packets for 3 levels decomposition. Node indices are written inside the boxes that represent the nodes in the wavelet tree decomposition. As shown in figure 5 that node 7 and node 6 are too far in frequency while they are subsequent nodes as wavelet packets indexing system. This problem needs to be altered in such that adjacent frequency bands are listed as contiguous numbers. This way we will ensure that indexing system reflects frequency scale. This property may be used in the scoring system. Information in figure 5 is tabulated in table I to make it simple to figure out adjacent bands. Traversing tree as Left -> Right -> Center will be very logical to make good criteria for adjacency. Figure 6 explains the new indexing system.

Now we are ready to apply the best tree algorithm to optimize the full binary tree shown in figure 6. The optimization minimizes the number of tree nodes such that it best fit the information included in the speech signal. The entropy is used in the optimization algorithm.

Fig 6. Proposed Indexing to Solve the Adjacency Problem due to Wavelet Packet’s Indexing System.

The number is formed such that it reflects the tree structure within the cluster. Trees that cover the same bands will be almost adjacent trees. This property will be utilized in the scoring system. By considering all clusters, a vector of 4 components will be formed. Each vector’s component represents a certain cluster. And each cluster covers a certain area in the total bandwidth. This is the 4 point encoded method that construct BTE features vector.

Figure 7 introduces a simple example to explain features encoding for a frame of speech signal. Circles in figure 7 represent leave nodes in the best tree decomposition.

Fig 7. Proposed Indexing to Solve the Adjacency Problem due to Wavelet Packet’s Indexing System.
The indicated tree structure in figure 7 will be encoded into features vector of 4 elements as shown in table II.

<table>
<thead>
<tr>
<th>Element</th>
<th>Binary Value</th>
<th>Decimal Value</th>
<th>Frequency Band</th>
</tr>
</thead>
<tbody>
<tr>
<td>V1</td>
<td>0001100</td>
<td>12</td>
<td>0 - 25 %</td>
</tr>
<tr>
<td>V2</td>
<td>1000000</td>
<td>64</td>
<td>25% - 50%</td>
</tr>
<tr>
<td>V3</td>
<td>0000000</td>
<td>0</td>
<td>50% - 75%</td>
</tr>
<tr>
<td>V4</td>
<td>0000100</td>
<td>4</td>
<td>75% - 100%</td>
</tr>
</tbody>
</table>

IV. LINEAR PREDICTION CODING

In modern signal processing, the analysis procedure extracts useful information from the structure of a signal. LPC technique is a developed algorithm used in speech analysis for many years. Its basic idea comes from a model representing the resonances of the human vocal tract. In general, speech sounds are produced by acoustic excitation of the vocal tract. During the production of voiced sounds, the vocal tract is excited by a series of nearly periodic pulses generated by the vocal cords. With unvoiced sounds, the excitation is provided by air passing turbulently through constrictions in the tract [13-16]. Fig. 1 shows the speech signal production model in which the speech synthesizer strongly depends on the estimation of \( a_p \). Here, \( a_p \) are autoregressive parameters obtained from the linear prediction method, and provide better results to characterize human speech. The use of these parameters assumes the speech signal can be represented as the output signal of an all pole digital filter in which the excitation is an impulse sequence with a frequency equal to the pitch of the speech signal under analysis when the segment is voiced, or with noise when the segment is unvoiced [17-19].

\[
\hat{s}(n) = -(a_1 s(n-1) + a_2 s(n-2) + \ldots + a_p(n-p)) \tag{9}
\]

Or

\[
\hat{s}(n) = -\sum_{k=1}^{p} a_k s(n-k), \quad k = 1, 2, \ldots, p. \tag{10}
\]

Therefore, it is affirmed a filter can be designed to estimate the data at time \( n \) only using the previous data at time \( n-1 \):

\[
\hat{s}(n) = -a_1s(n-1), \tag{11}
\]

V. RESULT

The various speech signal sample is simulated using MATLAB. The speech signal is segmented into packets with 640 samples. The Wavelet Packet Transform decomposition is computed for each packet. The transformed coefficients are extracted for further processing and the energy of the input signal is computed.

The results obtained for SNR, PSNR, NRMSE are calculated using the formulas:-

1) Signal to Noise Ratio:
This value gives the quality of reconstructed signal. Higher the value, the better:

\[
\text{SNR} = 10 \log_{10} \left( \frac{\sigma_x^2}{\sigma_e^2} \right) \tag{12}
\]

\( \sigma_x^2 \) is the mean square of the speech signal and \( \sigma_e^2 \) is the mean square difference between the original and reconstructed signals.

2) Mean Square Error:

\[
\text{MSE} = \frac{1}{N} \sum_{i=0}^{N-1} (x_i - r_i)^2 \tag{13}
\]

Where \( x_i \) is the original speech signal data and \( r_i \) is the reconstructed signal and \( N \) is the length of the reconstructed signal.

3) Peak Signal to Noise Ratio:

\[
\text{PSNR} = 10 \log_{10} \left[ \frac{NX^2}{\|x - r\|^2} \right] \tag{14}
\]

Where \( X \) is the maximum absolute square value of the signal \( x \).
4) **Compression Ratio (CR):**
   It is the ratio of length of the original signal to the compressed signal.
   \[
   CR = \frac{\text{length}(x(n))}{\text{Length}(cr)}
   \]
   (15)
   Where \(cr\) is the length of the compressed vector.

<table>
<thead>
<tr>
<th>TABLE III</th>
<th>PERFORMANCE FOR MALE SPOKEN.</th>
</tr>
</thead>
<tbody>
<tr>
<td>CR</td>
<td>SNR</td>
</tr>
<tr>
<td>2.5515</td>
<td>-0.9807</td>
</tr>
<tr>
<td>3.1862</td>
<td>-0.9632</td>
</tr>
<tr>
<td>4.2412</td>
<td>-0.9399</td>
</tr>
<tr>
<td>6.3409</td>
<td>-0.8993</td>
</tr>
<tr>
<td>12.5574</td>
<td>-0.8284</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>TABLE IV</th>
<th>PERFORMANCE FOR FEMALE SPOKEN.</th>
</tr>
</thead>
<tbody>
<tr>
<td>CR</td>
<td>SNR</td>
</tr>
<tr>
<td>2.5515</td>
<td>-4.6337</td>
</tr>
<tr>
<td>3.1862</td>
<td>-4.5778</td>
</tr>
<tr>
<td>4.2412</td>
<td>-4.5078</td>
</tr>
<tr>
<td>6.3409</td>
<td>-4.4042</td>
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<td>12.5574</td>
<td>-4.1998</td>
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</table>

<table>
<thead>
<tr>
<th>TABLE V</th>
<th>PERFORMANCE FOR MUSIC.</th>
</tr>
</thead>
<tbody>
<tr>
<td>CR</td>
<td>SNR</td>
</tr>
<tr>
<td>2.5515</td>
<td>-1.8229</td>
</tr>
<tr>
<td>3.1862</td>
<td>-1.7755</td>
</tr>
<tr>
<td>4.2412</td>
<td>-1.6934</td>
</tr>
<tr>
<td>6.3409</td>
<td>-1.6274</td>
</tr>
<tr>
<td>12.5574</td>
<td>-1.4804</td>
</tr>
</tbody>
</table>

**VI. CONCLUSION**

Wavelet packets make a similar processing on speech signal as the Filter banks method. It is much smarter than filter banks in that the number of filters is adapted by considering signal entropy to find the best tree. The problem of having dynamic size feature vectors is solved by considering the 4 points encoding algorithm. The proposed encoding system guarantees that minimizing distance between feature vectors based on adjacency in frequency domain. This adjacency based on frequency domain of feature vectors distance calculation makes (BTE) features are highly promising in speech compression systems.

**REFERENCES**

Amr M. Gody received the B.Sc. M.Sc., and PhD. from the Faculty of Engineering, Cairo University, Egypt, in 1991, 1995 and 1999 respectively. He joined the teaching staff of the Electrical Engineering Department, Faculty of Engineering, Fayoum University, Egypt in 1994. He is a co-author of about 50 papers in national and international conference proceedings and journals. He is the Acting chief of Electrical Engineering department, Fayoum University in 2010, 2012 till now. His current research areas of interest include speech processing, speech recognition and speech compression.

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